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Docket No. 02100033aa In re: ✓ patent/___ trademark application of

Applicant(s) K. Ozawa

Serial No. 09/302,397 Date Filed 4/30/99

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re patent application of

Kazunori Ozawa

Serial No. 09/302,397 | Group Art Unit 2654

Filed April 30, 1999 | Examiner Angela A. Armstrong

For SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS

Box Non-Fee Amendment
Assistant Commissioner for Patents
Washington, D.C. 20231

AMENDMENT UNDER 37 C.F.R. §1.111

Sir:

In response to the Office Action mailed April 2, 2002, please amend the above-identified patent application as follows:

In the Specification:

Please amend the specification as follows. A clean copy of the amended paragraphs is attached.

Please amend the paragraph beginning on page 2, line 22, and continuing to page 3, line 3, as follows:

This arises from the fact that, in the methods in references 1 and 2, in order to select a sound source code vector, filtering or convolution calculation is performed once for each code [vectors] <u>vector</u>, and such calculation is repeated by a number of times equal to the number of code vectors stored in the codebook.

Please amend the paragraph on page 10, lines 6 to 18, as follows:

As is obvious from the above aspects, according to the present invention, the mode is discriminated on the basis of the past quantized gain of the adaptive codebook. If a predetermined mode is discriminated, combinations of code vectors stored in the codebook, which [is] <u>are</u> used to collectively quantize the amplitude or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions are searched to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. With this arrangement, even if the bit rate is low, a background noise portion can be properly coded with a relatively small [amount] calculation amount.

Please amend the paragraph on page 12, lines 2 to 18, as follows:

Several embodiments of the present invention will be described below with reference to the accompanying drawings. In a speech coding apparatus according to an embodiment of the present invention, a mode discrimination circuit (370 in Fig. 1) discriminates the mode on the basis of the past quantized gain of an adaptive codebook. When a predetermined mode is discriminated, a sound source quantization circuit (350 in Fig. 1) searches combinations of code vectors stored in a codebook (351 or 352 in Fig. 1), which is used to collectively quantize the amplitudes or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions, to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. A gain quantization circuit ([365] <u>366</u> in Fig. 1) quantizes gains by using a gain codebook (380 in Fig. 1).

Please amend the paragraph beginning on page 17, line 19, and continuing to page 18, line 5, as follows:

For example, linear predictive coefficients calculated for the second and fourth subframes based on the Burg method are transformed into LSP paramete3rs

whereas LSP parameters for the first and third subframes are determined by linear interpolation, and the LSP parameters of the first and third subframes are inversely transformed into linear predictive coefficients. Then, the linear predictive coefficients α il (i=1, . . ., 10, [1] l=1, . . .,5) of the first to fourth subframes are output to a perceptual weighting circuit 230. The LSP parameters of the fourth subframe are output to the spectrum parameter quantization circuit 210.

Please amend the paragraph on page 18, lines 6 to 15, as follows:

The spectrum parameter quantization circuit 210 efficiently quantizes the LSP parameters of a predetermined subframe from the spectrum parameters and outputs a quantization value which minimizes the distortion given by:

$$D_j = \sum_{i=1}^{p} W(i)[LSP(i) - QLSP(i)_j]^2 \qquad \ldots(1)$$

where LSP(i), $QLSP(i)_j$, and W(i) are the LSP [parameter] parameters of the ith-order before quantization, the jth result after the quantization, and the weighting coefficient, respectively.

Please amend the paragraph on page 20, lines 6 to 15, as follows:

The LSP parameters of the first to third subframes reconstructed in such a manner as described above and the quantization LSP parameters of the fourth subframe are transformed into linear predictive coefficients α il (i=1, . . ., 10, [1] l=1, . . .,5) for each subframe, and the linear predictive coefficients are output to the impulse response calculation circuit 310. Furthermore, an index representing the code vector of the quantization LSP parameters of the fourth subframe is output to a multiplexer 400.

Please amend the paragraph on page 20, lines 16 to 22, as follows:

The perceptual weighting circuit 230 receives the linear predictive coefficients α il (i=1, . . ., 10, [1] l=1, . . .,5) before quantization for each subframe

from the spectrum parameter calculation circuit 200, performs perceptual weighting for the speech signal of the subframe on the basis of the method described in reference 1 and outputs a resultant preceptual weighting signal.

Please amend the paragraph beginning on page 22, line 13, and continuing to page 24, line 2, as follows:

The adaptive codebook circuit 500 receives a sound source signal v(n) in the past from a gain quantization circuit 366, receives the output signal $x'_w(n)$ from the subtractor 235 and the impulse responses $h_w(n)$ from the impulse response calculation circuit 310. Then, the adaptive codebook circuit 500 calculates a delay [DT] $\underline{D_T}$ corresponding to pitch, which minimizes the distortion given by:

$$D_T = \sum_{n=0}^{N-1} x'^2_w(n) - \frac{\left[\sum_{n=0}^{N-1} x'_w(n)\, y_w(n-T)\right]^2}{\left[\sum_{n-0}^{N-1} y_w^2(n-T)\right]} \qquad \ldots(7)$$

$$\text{for } y_w(n-T) = v(n-T)*h_w(n) \qquad \ldots(8)$$

and outputs an index representing the delay to the multiplexer 400[.], where the symbol * signifies a convolution calculation.

Please amend the paragraph on page 24, lines 10 to 16, as follows:

For a voiced sound, a B-bit amplitude codebook or polarity codebook is used to collectively quantize the amplitudes of pules in units of M pulses. A case wherein the polarity codebook is used will be described below. This polarity codebook is stored in a codebook 351 for a voiced sound, and is [store din] <u>stored in</u> a codebook 352 for an unvoiced sound.

Please amend the paragraph beginning on page 24, line 24, and continuing to page 25, line 4, as follows:

Equation (11) can be minimized by obtaining a combination of an amplitude code vector k and a position [mi] m_i which maximizes $D_{(k, i)}$ given by:

$$D_{(k, j)} = \frac{\left[\sum_{n=0}^{N-1} e_w(n)\, s_{wk}(m_i)\right]^2}{\sum_{n=0}^{N-1} s_{wk}^2(m_i)} \qquad \ldots(12)$$

where [s_{wk}(mi)] $s_{wk}(m_i)$ is calculated according to equation (5) above.

Please amend the paragraph on page 32, lines 7 to 14, as follows:

Referring to Fig. [15] 5, in the fifth embodiment of the present invention, a demultiplexer section 510 demultiplexes a code sequence input through an input terminal 500 into a spectrum parameter, an adaptive codebook delay, an adaptive codebook vector, a sound source gain, an amblitude or polarity code vector as sound source information, and a code representing a pulse position, and outputs them.

In the Claims:

Please amend claims 6 and 7 as follows. A clean copy of the amended claims is attached.

Claim 6 (Twice Amended). A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to [poutout] output a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including[:]

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codeboook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized

sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

Claim 7 (Twice Amended). A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of [a] an adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section,

said sound source quantization section [for] outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output

from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which includes spectrum parameters and reproduces a speech signal by filtering the sound source signal.

Claim 8 (Amended). A speech coding apparatus comprising:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter;

means for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal; and

mode discrimination means for receiving a past quantized adaptive codebook gain and [performs] performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold, and

further comprising:

sound source quantization means for quantizing a sound source

signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally [shifting] shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech;

gain quantization means for quantizing a gain by using a gain codebook; and

multiplex means for outputting a combination of outputs from said spectrum parameter calculation means, said adaptive codebook means, said sound source quantization means, and said gain quantization means.

REMARKS

The specification has been amended to correct minor typographical and grammatical errors. No new matter has been added.

Claims 1 to 11 remain in the application. Claims 6, 7 and 8 have been amended.

Claims 1 to 11 were rejected under 35 U.S.C. §112, first paragraph, as containing subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. The Examiner states that claims 1 to 11, as argued by applicant, include subject matter for processing pulses by using different pulse-shifting schemes depending upon whether a voice sound mode or an unvoiced sound mode is discriminated. The Examiner contends that the specification does not support a time shift of both voiced and unvoiced sound modes. The rejection is respectfully traversed.

The Examiner refers to page 29, lines 14 to 19, of the specification, saying that the time shift amount is indicated by $\delta(j)$. It appears that the Examiner's page

citation is wrong. δ(j) is defined on page 26, line 18. The Examiner takes the position that the time shifts are used only in minimizing distortions of unvoiced signals and, therefore, an argument that time shifting is employed for minimizing distortions in both voiced and unvoiced signals is outside the scope of the disclosed invention. First of all, the disclosure stands on its own, and any arguments, however interpreted by the Examiner, are not part of the disclosure. So from that point of view, the rejection is improper. Second, the Examiner has quite clearly not understood the invention or the relevance of the equations she has cited as supporting her position that there is no time shifting in voiced mode. In the paragraphs preceding equation 11 on page 24, Applicant explains that an amplitude codebook or a polarity codebook can be used. The case of a polarity codebook is described wherein codebook 351 is used for voiced sound and codebook 352 is used for unvoiced sound. The point here is that Applicant has used separate codebooks for voiced and unvoiced speech signals so as to minimize calculations without adversely affecting sound quality in terms of background noise.

Since the claims are clearly supported by the specification and drawings as originally filed, the rejection under 35 U.S.C. §112, first paragraph, is without ground and should be withdrawn.

Claims 1 to 11 were additionally rejected under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent No. 5,704,003 to Kleijn et al. in view of U.S. Patent No. 5,751,903 to Swaminathan et al. and U.S. Patent No. 5,657,418 to Gershon et al. This rejection is respectfully traversed for the reason that the combination of references relied on by the Examiner fails to show or suggest the claimed invention.

The claimed invention is directed to a speech coding apparatus for coding a speech signal at a low bit rate with high quality. The invention effectively suppresses deterioration in sound quality in terms of background noise while minimizing the calculations required. The speech coding apparatus uses a mode discrimination circuit 370 (Fig. 1) which discriminates the mode on the basis of

the past quantized gain of an adaptive codebook. The past quantized gain is shown in Figure 1 by the input to the mode discrimination circuit 370 from gain quantization circuit 366, and the adaptive codebook 500 also receives an input from gain quantization circuit 366. When a predetermined mode is discriminated as either voiced or unvoiced, a sound source quantization circuit 350 searches combinations of code vectors stored in a code book 351 or 352, which is used to collectively quantize the amplitudes or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions, to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. The gain quantization circuit 366 quantizes gains by using a gain codebook 380.

A main characteristic feature of the present invention as claimed in claim 1 is that a speech coding apparatus comprises a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from the discrimination section indicates a predetermined mode, and searches combinations of code vectors stored in the codebook and a plurality of shift amounts used to shift positions of the pulse so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech.

Specifically, claim 1 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". This sound source quantization section comprises, "a discrimination section [**370**] for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a *past quantized gain* of an adaptive codebook [**500**]" (emphasis added), and "a sound source quantization section [**350**] which has a codebook [**351, 352**] for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination circuit section, and *searches combinations of code vectors stored in*

said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech" (emphasis added).

Claim 1 is directed to the first embodiment shown in Figure 1 of the drawings. Claims 2, 3 , 4, and 5 are directed respectively to the second, third, fourth, and fifth embodiments shown in Figures 2, 3, 4, and 5 of the drawings and contain similar limitations.

Specifically, claim 2 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section **[370]** for discriminating a voice sound mode and an unvoiced sound mode *on a basis of a past quantized gain of an adaptive codebook*" (emphasis added), and "a sound source quantization section **[350]** which has a codebook **[351, 352]** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and *outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule*" (emphasis added). Note that in the embodiment of Figure 2, a random number generating circuit 600 is added to generate a predetermined number of pulse positions, as described in more detail beginning at page 29, line 23, and continuing to page 30, line 4.

Claim 3 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section **[370]** for discriminating a voice sound mode and an unvoiced sound mode *on the basis of a past quantized gain of an adaptive codebook* **[500]**" (emphasis added), and "a sound source quantization section **[350]** which has a codebook **[351, 352]** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing

amplitudes or polarities of the pulses based on an output from said discrimination section, and a gain codebook [**380**] for quantizing gains, and *searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech*" (emphasis added).

Claim 4 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section [**370**] for discriminating a voice sound mode and an unvoiced sound mode *on the basis of a past quantized gain of a adaptive codebook* [**500**]" (emphasis added), and "a sound source quantization section [**350**] which has a codebook [**351, 352**] for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on output from said discrimination section, and a gain codebook [**380**] for quantizing gains, and *outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule*" (emphasis added).

Claim 5 is directed to the decoder apparatus and recites, "a mode discrimination section [**530**] for discriminating a voice sound mode and an unvoiced sound mode by *using a past quantized gain in said adaptive codebook* [**520**]" (emphasis added), and "*a sound source signal reconstructing section* [**540**] *for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information based on an output from said discrimination section*, wherein a speech signal is reproduced by passing the sound source signal through a synthesis filter section [**560**] which includes spectrum parameters" (emphasis added).

Claims 6 and 7 recite the combination of a speech coding and decoding apparatus and, again, recite limitations similar to those pointed out above. Claim 8 is directed to a speech coding apparatus which comprises, *inter alia*, "sound source quantization means for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook *for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech*" (emphasis added). Claims 9, 10 and 11 are dependent on claim 8.

In making the rejection, the Examiner acknowledges that Kleijn et al. do not teach discriminating a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook. The Examiner cited Gerson et al. for a teaching of that a lag parameter, which reflects the periodicity, is used to select a particular coding mode. From this, the Examiner concludes that it would have been obvious to modify Kleijn et al. to implement discriminating a voiced/unvoiced mode based on past quantized gain. What the Examiner is attempting to do is to ignore the clear teaching of the primary reference and attribute to the secondary reference a teaching which is clearly not warranted by the reference. The combination she proposes is based solely on hindsight not permitted by Section 103 of the Patent Statute.

The Examiner further takes the position that Kleijn et al. teach sound source quantization by searching a codebook for code vectors and delays so as to output a combination of code vector and shift amount that minimizes distortion. This is not an accurate characterization of the reference. A characteristic feature of Kleijn et al. is that residual signals are coded by a time shift. That is, as described in column 6, after line 14, the best value for time shift T which can minimize an error output between a signal r(n-T) obtained by shifting the residual signal r(n) by T and a delayed residual signal r(n-D(n)) is required, whereby the parameter

required in coding is selected. The Examiner goes on to acknowledge that Kleijn et al. do not teach a multiplexer for the coder or decoder scheme, but cites Swaminathan et al., saying that "it would have been obvious to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as *suggested* by Swaminathan et al." (emphasis added). This, again, is a wholesale reconstruction of the primary reference unwarranted by what the references in fact teach.

The Examiner is reminded of the basic considerations which apply to obviousness rejections as set out in MPEP 2141. Specifically, "When applying 35 U.S.C. 103, the following tenets of patent law must be adhered to:

"(A) The claimed invention must be considered as a whole;

"(B) The references must be considered as a whole and must suggest the desirability and thus the obviousness of making the combination;

"(C) The references must be viewed without the benefit of impermissible hindsight vision afforded by the claimed invention; and

"(D) Reasonable expectation of success is the standard with which obviousness is determined."

The Examiner has taken three rather diverse systems and tried to combine them based on Applicant's own disclosure. It is not even clear that the reconstructions she proposes would result in an operable system, particularly since the references are each based on different principles of operation. The rejection is clearly without merit and should therefore be withdrawn.

In view of the foregoing, it is respectfully requested that the application be reconsidered, that claims 1 to 11 be allowed, and that the application be passed to issue.

Should the Examiner find the application to be other than in condition for allowance, the Examiner is requested to contact the undersigned at the local telephone number listed below to discuss any other changes deemed necessary in a telephonic or personal interview.

A provisional petition is hereby made for any extension of time necessary for the continued pendency during the life of this application. Please charge any fees for such provisional petition and any deficiencies in fees and credit any overpayment of fees to Attorney's Deposit Account No. 50-2041.

Respectfully submitted,

C. Lamont Whitham
Reg. No. 22,424

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Clean Copy of Amended Paragraphs

Paragraph beginning on page 2, line 22, and continuing to page 3, line 3, now reads as follows:

C1

This arises from the fact that, in the methods in references 1 and 2, in order to select a sound source code vector, filtering or convolution calculation is performed once for each code vector, and such calculation is repeated by a number of times equal to the number of code vectors stored in the codebook.

Paragraph on page 10, lines 6 to 18, now reads as follows:

C2

As is obvious from the above aspects, according to the present invention, the mode is discriminated on the basis of the past quantized gain of the adaptive codebook. If a predetermined mode is discriminated, combinations of code vectors stored in the codebook, which are used to collectively quantize the amplitude or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions are searched to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. With this arrangement, even if the bit rate is low, a background noise portion can be properly coded with a relatively small calculation amount.

Paragraph on page 12, lines 2 to 18, now reads as follows:

C3

Several embodiments of the present invention will be described below with reference to the accompanying drawings. In a speech coding apparatus according to an embodiment of the present invention, a mode discrimination circuit (370 in Fig. 1) discriminates the mode on the basis of the past quantized gain of an adaptive codebook. When a predetermined mode is discriminated, a sound source quantization circuit (350 in Fig. 1) searches combinations of code vectors stored in a codebook (351 or 352 in Fig. 1), which is used to collectively quantize the amplitudes or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions, to select a

C3 cont

combination of a code vector and shift amount which minimizes distortion relative to input speech. A gain quantization circuit (366 in Fig. 1) quantizes gains by using a gain codebook (380 in Fig. 1).

Paragraph beginning on page 17, line 19, and continuing to page 18, line 5, now reads as follows:

C4

For example, linear predictive coefficients calculated for the second and fourth subframes based on the Burg method are transformed into LSP paramete3rs whereas LSP parameters for the first and third subframes are determined by linear interpolation, and the LSP parameters of the first and third subframes are inversely transformed into linear predictive coefficients. Then, the linear predictive coefficients α il (i=1, . . ., 10, l=1, . . .,5) of the first to fourth subframes are output to a perceptual weighting circuit 230. The LSP parameters of the fourth subframe are output to the spectrum parameter quantization circuit 210.

Paragraph on page 18, lines 6 to 15, now reads as follows:

C5

The spectrum parameter quantization circuit 210 efficiently quantizes the LSP parameters of a predetermined subframe from the spectrum parameters and outputs a quantization value which minimizes the distortion given by:

$$D_j = \sum_{i=1}^{P} W(i)[LSP(i) - QLSP(i)_j]^2 \qquad \ldots(1)$$

where LSP(i), $QLSP(i)_j$, and W(i) are the LSP parameters of the ith-order before quantization, the jth result after the quantization, and the weighting coefficient, respectively.

Paragraph on page 20, lines 6 to 15, now reads as follows:

C6

The LSP parameters of the first to third subframes reconstructed in such a manner as described above and the quantization LSP parameters of the fourth subframe are transformed into linear predictive coefficients α il (i=1, . . ., 10, l=1,

C6 cont

. . .,5) for each subframe, and the linear predictive coefficients are output to the impulse response calculation circuit 310. Furthermore, an index representing the code vector of the quantization LSP parameters of the fourth subframe is output to a multiplexer 400.

Paragraph on page 20, lines 16 to 22, now reads as follows:

C7

The perceptual weighting circuit 230 receives the linear predictive coefficients α il (i=1, . . ., 10, l=1, . . .,5) before quantization for each subframe from the spectrum parameter calculation circuit 200, performs perceptual weighting for the speech signal of the subframe on the basis of the method described in reference 1 and outputs a resultant preceptual weighting signal.

Paragraph beginning on page 22, line 13, and continuing to page 24, line 2, now reads as follows:

C8

The adaptive codebook circuit 500 receives a sound source signal v(n) in the past from a gain quantization circuit 366, receives the output signal $x'_w(n)$ from the subtractor 235 and the impulse responses $h_w(n)$ from the impulse response calculation circuit 310. Then, the adaptive codebook circuit 500 calculates a delay D_T corresponding to pitch, which minimizes the distortion given by:

$$D_T = \sum_{n=0}^{N-1} x'^2_w(n) - \frac{\left[\sum_{n=0}^{N-1} x'_w(n)\, y_w(n-T)\right]^2}{\left[\sum_{n-0}^{N-1} y^2_w(n-T)\right]} \quad \ldots(7)$$

$$\text{for } y_w(n-T) = v(n-T)*h_w(n) \quad \ldots(8)$$

and outputs an index representing the delay to the multiplexer 400, where the symbol * signifies a convolution calculation.

Paragraph on page 24, lines 10 to 16, now reads as follows:

C9

For a voiced sound, a B-bit amplitude codebook or polarity codebook is used to collectively quantize the amplitudes of pules in units of M pulses. A case wherein the polarity codebook is used will be described below. This polarity codebook is stored in a codebook 351 for a voiced sound, and is stored in a codebook 352 for an unvoiced sound.

Paragraph beginning on page 24, line 24, and continuing to page 25, line 2, now reads as follows:

C10

Equation (11) can be minimized by obtaining a combination of an amplitude code vector k and a position m_i which maximizes $D_{(k,\ i)}$ given by:

$$D_{(k,\ j)} = \frac{\left[\sum_{n=0}^{N-1} e_w\ (n)\ s_{wk}\ (m_i)\right]^2}{\sum_{n=0}^{N-1} s_{wk}^2\ (m_i)} \qquad \ldots(12)$$

where $s_{wk}(m_i)$ is calculated according to equation (5) above.

Paragraph on page 32, lines 7 to 14, now reads as follows:

C11

Referring to Fig. 5, in the fifth embodiment of the present invention, a demultiplexer section 510 demultiplexes a code sequence input through an input terminal 500 into a spectrum parameter, an adaptive codebook delay, an adaptive codebook vector, a sound source gain, an amblitude or polarity code vector as sound source information, and a code representing a pulse position, and outputs them.

(TWICE AMENDED)

6. A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

C12

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codeboook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

(TWICE AMENDED)

C12 cont

7. A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said

discrimination section,

said sound source quantization section outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

C12 cont

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which includes spectrum parameters and reproduces a speech signal by filtering the sound source signal.

(TWICE AMENDED)

8. A speech coding apparatus comprising:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter;

means for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal; and

mode discrimination means for receiving a past quantized adaptive codebook gain and performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold, and

further comprising:

C 12 cont

sound source quantization means for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech;

gain quantization means for quantizing a gain by using a gain codebook; and

multiplex means for outputting a combination of outputs from said spectrum parameter calculation means, said adaptive codebook means, said sound source quantization means, and said gain quantization means.